IN THE CLAIMS

Please amend the claims as follows:

1. (Currently Amended) A method for controlling a Voice ~~Over~~ over Internet Protocol (VoIP) call at a telephone endpoint, comprising:

tracking adaptation ~~schemes~~ scheme settings used at the telephone endpoint for transmitting packets in the VoIP call, the telephone endpoint being an originating source for the VoIP call;

accessing the adaptation scheme settings to determine a current preferred encoding algorithm for encoding an audio input for the VoIP call at the telephone endpoint;

encoding a portion of the audio input using the current preferred encoding algorithm and then transmitting the encoded portion of the audio input from the telephone endpoint;

monitoring a user response to the VoIP call that requests a different level of user perceived sound quality for the VoIP call; and

selecting a new preferred encoding algorithm to vary the user perceived sound quality;

updating the adaptation scheme settings with the new preferred encoding algorithm selection during the VoIP call; and

encoding a remaining portion of the audio input at the telephone endpoint using the preferred new encoding algorithm and then transmitting the encoded remaining portion of the audio input from the telephone endpoint.

~~dynamically varying the adaptation schemes used at the telephone endpoint for transmitting the packets in the VoIP call from the telephone endpoint to correspond with the requested level of user perceived sound quality;~~

~~wherein dynamically varying the adaptation schemes affects how much digital data is used to represent an audio signal.~~

2. (Previously Presented) A method according to claim 1 including;

initially transmitting the packets in the VoIP call over an Internet Protocol (IP) packet switched network using an IP packet best effort transmission scheme;

monitoring the user response for a request to increase sound quality; and

requesting reservation of IP packet switched network resources during the already established VoIP call when the increase sound quality request is detected from the user response prior to the reserved IP packet switched network resources being used during the VoIP call and without necessarily using the entire requested resources during the VoIP call.

3. (Previously Presented) A method according to claim 2 wherein requesting reservation of network resources comprises making an RSVP (Resource Reservation Protocol) request during the VoIP call.

4. (Original) A method according to claim 2 including conducting the already established VoIP call using reserved network resources when the requested reservation is accepted and the user response requests additional increases in the sound quality of the VoIP call.

5. (Original) A method according to claim 4 including increasing voice coder performance or reducing payload size after the network resources are reserved.

6. (Original) A method according to claim 1 including using a signal generated by an input device to detect the user response during the VoIP call.

7. (Original) A method according to claim 6 including using a dial or buttons on a telephone as the input device.

8. (Original) A method according to claim 6 including using a graphical user interface as the input device.

9. (Original) A method according to claim 1 including decoding Dual Tone Multiple Frequency signals to detect the user response.

10. (Currently Amended) A method for controlling a VoIP call, comprising:

tracking adaptation schemes used for transmitting packets in a Voice ~~Over~~ <u>over</u> IP (VoIP) call;

monitoring a user response to the VoIP call;

dynamically varying the adaptation schemes used for transmitting the packets according to a user adaptation scheme selection during the VoIP call ~~the monitored user response~~; and

monitoring congestion in a network used for conducting the VoIP call; ~~and~~

varying the adaptation schemes according to both the ~~user response~~ user adaptation scheme selection and the monitored congestion;

wherein the user adaptation scheme selection ~~dynamically varying the adaptation schemes includes either varying which~~ controls which coder algorithm is used at the telephone endpoint, ~~varying~~ a packet payload size of the packets or ~~varying~~ what type of Forward Error Correction (FEC) is used in association with the packets.

11. (Currently Amended) A method according to claim [[1]] 10 further including automatically updating a network resource reservation in response to the user adaptation scheme selection. ~~wherein varying the adaptation schemes comprises varying codecs used for encoding audio signals into digital data making up the packets~~.

12. (Currently Amended) A method according to claim [[1]] 10 further including:

receiving a user cost selection for the VoIP call; and

automatically varying the adaptation schemes in response to the user cost selection for the VoIP call. ~~including detecting a user response selecting a cost for the VoIP call and varying the adaptation schemes according to the selected cost.~~

13. (Currently Amended) An adaptation system, comprising:

an input ~~for detecting~~ to detect a user response requesting a different user perceived sound quality for a call and specifying a user-defined codec selection; and

a controller configured to measure current network congestion of a network that transmits the call and determine a packet loss ratio associated with the requested different user perceived sound quality, the user-defined codec selection and the measured current network congestion;

the controller to compare the determined packet loss ratio to a predetermined packet loss ratio that is set according to an empirical analysis identifying a threshold amount of packet loss that represents a minimum call sound quality;

the controller to dynamically vary ~~adaptation parameters~~ a codec used to encode the call while the call is in progress ~~used for transmitting packets making up the call to correspond with~~ according to the user-defined codec selection when the determined packet loss ratio is less than the predetermined packet loss ratio;

the controller to determine a system-defined codec selection corresponding to both the measured network congestion and the determined packet loss ratio when the determined packet loss ratio is equal to or greater than the predetermined packet loss ratio; and

the controller to dynamically vary the codec used to encode the call while the call is in progress according to the system-defined codec selection when the determined packet loss ratio is equal to or greater than the predetermined packet loss ratio ~~the requested different user perceived sound quality in the user response detected by the input~~;

~~wherein dynamically varying the adaptation parameters affects how an analog signal is converted into the packets making up the call.~~

14. (Cancelled)

15. (Currently Amended) An adaptation system according to claim 13 wherein the controller initially transmits [[the]] packets in the call using a best effort transmission scheme and during the call requests reservation of network resources when the user response requests increased sound quality.

16. (Previously Presented) An adaptation system according to claim 15 wherein the controller initiates an RSVP (Resource Reservation Protocol) request to reserve the network resources.

17. (Currently Amended) An adaptation system according to claim 15 wherein the controller monitors for acceptance of the network reservation request and modifies the ~~adaptation parameters~~ codec to provide an increased sound quality call ~~when~~ only after the acceptance is received.

18. (Original) An adaptation system according to claim 13 wherein the input comprises a dial or buttons.

19. (Original) An adaptation system according to claim 13 wherein the input comprises a graphical user interface.

20. (Currently Amended) An adaptation system according to claim 19 including a cost icon in the graphical user interface that allows selection of a call cost, the controller restricting the system-defined according selection according to the ~~varying the adaptation parameters according to the~~ selected call cost.

21. (Original) An adaptation system according to claim 13 wherein the input device generates Dual Tone Multiple Frequency signals that are decoded by the controller for identifying the user response.

22. (Currently Amended) An adaptation system according to claim 13 further comprising the controller modifying a packet payload size for the call in response to the requested different user perceived sound quality ~~wherein the user response determines how much the controller varies the adaptation parameters~~.

23. (Original) An adaptation system according to claim 13 wherein the controller varies a rate that the packets are transmitted and received during the call.

24. (Currently Amended) An electronic storage medium containing software used for controlling a Voice over IP (VoIP) call, the software in the electronic storage medium comprising:

code for ~~tracking adaptation schemes used~~ determining a current payload size for ~~transmitting~~ audio packets in [[a]] the Voice Over IP (VoIP) call;

code for monitoring a user response to the VoIP call indicating a desired level of user perceived audio quality for the VoIP call; ~~and~~

code for determining a current jitter amount for the VoIP call;

code for ~~dynamically varying the adaptation schemes~~ changing the current payload size for the audio packets mid-call to correspond with the desired level of user perceived audio quality only when the current jitter amount is less than a predetermined jitter amount that is preset and represents an acceptable perceivable sound quality. ~~used for transmitting the audio packets from a telephone endpoint so that the user perceived audio quality of the VoIP call corresponds with the monitored user response.~~

25. (Original) An electronic storage medium according to claim 24 including;

code for initially transmitting the packets in the VoIP call using a best effort transmission scheme;

code for monitoring the user response for a request to increase voice quality; and

code for requesting reservation of network resources during the already established VoIP call when the increase voice quality request is detected from the user response.

26. (Previously Presented) An electronic storage medium according to claim 25 including code that requests reservation of network resources by making an RSVP (Resource Reservation Protocol) request in the middle of the VoIP call.

27. (Original) An electronic storage medium according to claim 25 including code for conducting the already established VoIP call using reserved network resources when the requested reservation is accepted and the user response requests additional increases in voice quality of the VoIP call.

28. (Original) An electronic storage medium according to claim 27 including code for increasing voice coder quality and reducing packet payload size for the packets in the VoIP call after the network resources are reserved.

29. (Original) An electronic storage medium according to claim 24 including code that detects the user response from a signal generated by an input device controllable by a user during the VoIP call.

30. (Original) An electronic storage medium according to claim 29 wherein the input device comprises a dial on a telephone.

31. (Original) An electronic storage medium according to claim 29 wherein the input device comprises a graphical user interface on a computer.

32. (Original) An electronic storage medium according to claim 24 including code that decodes Dual Tone Multiple Frequency signals to identify the user response.

33. (Currently Amended) An electronic storage medium according to claim 24 including code for monitoring congestion in a network used for conducting the VoIP call and varying the ~~adaptation schemes~~ current payload size according to the user response and the monitored congestion.

34. (Currently Amended) An electronic storage medium according to claim 24 including:

code for varying codecs used for encoding audio signals into digital data making up ~~the~~ audio packets associated with the VoIP call data;

code for varying a rate that the audio packets are transmitted and received during the VoIP call;

code for varying an amount of audio data in the audio packets; and

code for adding or removing error correction information from the audio packets.

35. (Original) An electronic storage medium according to claim 24 including code for detecting a user response selecting a cost for the VoIP call and varying the adaptation schemes according to the selected cost.

36. (Currently Amended) A system for controlling a VoIP call, comprising:

means for tracking adaptation schemes used for transmitting audio packets in a Voice ~~Over~~ over IP (VoIP) call;

means for monitoring a user response to the VoIP call indicating a desired level of user perceived audio quality for the VoIP call; ~~and~~

means for dynamically varying the adaptation schemes used for transmitting the audio packets from a telephone endpoint in response to user indications so that the user perceived audio quality of the VoIP call corresponds with the monitored user response[[.]];

means for measuring a first packet loss rate for the VoIP call before dynamically varying the adaptation schemes;

means for measuring a second packet loss rate for the VoIP call after dynamically varying the adaptation schemes;

means for comparing the first packet loss rate to the second packet loss rate; and

means for automatically dynamically re-varying the adaptation schemes to lower bandwidth consumption in response to a determination that the second packet loss rate is a predetermined amount higher than the first packet loss rate.

37. (Currently Amended) A system for controlling a VoIP call, comprising:

means for tracking packet payload size for a VoIP call ~~tracking adaptation schemes used for transmitting audio packets in a Voice Over IP (VoIP) call~~;

means for monitoring a user response to the VoIP call indicating delays associated with the VoIP call;

means for determining current network congestion;

means for dynamically varying the ~~adaptation schemes used for transmitting the audio packets~~ packet payload size during the VoIP call so that packets of the VoIP call include payloads of varying sizes according to the monitored user response when the current network congestion is lower than a predetermined threshold ~~, dynamically varying the adaptation schemes changing how an analog audio signal is packetized into the audio packets~~;

means for initially transmitting the ~~packets in the VoIP call~~ the packets using a best effort transmission scheme;

means for monitoring the user response for a request to increase voice quality; and

means for requesting reservation of network resources for the call during the already established VoIP call when the increase voice quality request is detected from the user response and when the current network congestion is below a predetermined threshold[[.]]; and

means for indicating that the encoding process has not been dynamically varied when the current network congestion is not lower than a predetermined threshold.

38. (Previously Presented) A system according to claim 37 including means for requesting reservation of network resources by making an RSVP (Resource Reservation Protocol) request in the middle of the VoIP call.

39. (Original) A system according to claim 37 including means for conducting the already established VoIP call using reserved network resources when the requested reservation is accepted and the user response requests additional increases in voice quality of the VoIP call.

40. (Original) A system according to claim 38 including means for increasing voice coder quality and reducing packet payload size for the packets in the VoIP call after the network resources are reserved.

41. (Original) A system according to claim 36 including means for detecting the user response from a signal generated by an input device controllable by the user during the VoIP call.

42. (Original) A system according to claim 36 including means for detecting the user response from a dial on a telephone.

43. (Original) A system according to claim 36 including means for detecting the user response from a graphical user interface on a computer.

44. (Original) A system according to claim 36 including means for decoding Dual Tone Multiple Frequency signals to monitor the user response.

45. (Original) A system according to claim 36 including means for monitoring congestion in the network used for conducting the VoIP call and varying the adaptation schemes having a best chance with the monitored congestion of adapting the VoIP call to the user response.

46. (Original) A system according to claim 36 including:

means for varying codecs used for encoding audio signals into digital data making up the audio packets;

means for varying a rate that the audio packets are transmitted and received during the VoIP call;

means for varying an amount of audio data in the audio packets; and

means for adding or removing error correction information from the audio packets.

47. (Previously Presented) A system according to claim 36 including means for detecting a user response selecting a cost for the VoIP call and means for varying the adaptation schemes according to the selected cost.

48. (Currently Amended) A method for controlling a call, comprising;

~~establishing a call over a Plain Old Telephone System (POTS);~~

~~generating Dual Tone Multiple Frequency (DTMF) signals to request a controller to modify a sound quality of the call;~~

receiving a call over a circuit switched network;

packetizing a first portion of the call into first packets having a first packet payload size at a ~~network device~~ gateway that is coupled to a packet switched network;

receiving one or more Dual Tone Multiple Frequency (DTMF) tones over the circuit switched network from an origination source of the call, the one or more DTMF tones indicating a delay associated with the call;

packetizing a remaining portion of the call into second packets having a second packet payload size that is different than the first packet payload size in response to receiving the one or more DTMF tones.

~~modifying adaptation parameters response to the DTMF signals to modify the sound quality of the packetized call.~~

49. (Previously Presented) A method for controlling a Voice Over Internet Protocol (VoIP) call, comprising:

tracking adaptation schemes used for transmitting packets in the VoIP call;

monitoring a user response to the VoIP call that requests a different level of user perceived sound quality for the VoIP call; and

dynamically varying the adaptation schemes used for transmitting the packets in the VoIP call to correspond with the requested level of user perceived sound quality;

wherein dynamically varying the adaptation schemes includes adjusting Forward Error Correction (FEC) and adjusting packet payload length.

50. (Previously Presented) The method of claim 1 further comprising:

listening to an audible signal after dynamically varying the adaptation scheme to determine a level of user perceived sound quality for the audible signal;

further dynamically varying the adaptation schemes to improve the audible signal when the user perceived quality of the audible signal is low; and

further listening to the improved audible signal to determine a level of user perceived sound quality for the improved audible signal.

51. (New) The method of claim 49 wherein the packet payload length is dynamically varied according to measured network congestion.

52. (New) The method of claim 51 wherein the packet payload length is adjusted in response to a user indication of delays during the VoIP call.

53. (New) The method of claim 52 wherein the packet payload length is increased in response to the user indication.

54. (New) The method of claim 53 wherein the packet payload length is associated with a first packet payload type having 20 bytes when the delay is less than a predetermined threshold and the packet payload length is associated with a second packet payload type having at least 40 bytes when the delay is greater than the predetermined threshold.

55. (New) The method of claim 53 further comprising optimizing packet payload length according to a received maximum call cost selection.

56. (New) The method of claim 1 further comprising:

receiving a maximum call cost selection for the VoIP call indicating how much a user is willing to pay for the VoIP call; and

selecting the new preferred encoding algorithm according to the maximum call cost selection in addition to the user request.

57. (New) The method of claim 48 further comprising:

receiving a communication over the circuit switched network, the communication indicating how much a user is willing to pay for a portion of the VoIP call that is transferred over the packet switched network;

determining whether packetizing the remaining portion using the second packet payload size will cause the portion of the VoIP call that is transferred over the packet switched network to exceed an amount included in the indication;

determining a third packet payload size when the remaining portion using the second packet payload size will cause the portion of the VoIP call that is transferred over the packet switched network to exceed an amount included in the indication; and

packetizing the remaining portion of the call into second packets having the third packet payload size when the remaining portion using the second packet payload size will cause the portion of the VoIP call that is transferred over the packet switched network to exceed an amount included in the indication.